AETA AUDIO SYSTEMS

SCOOPY+

Mono / Stéréo portable audio codec
ISDN / POTS / GSM 3G+ / IP

User manual

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1. Scoopy+ - Easy getting started

To establish a Link over ISDN or POTS :

Connect the suitable cable connectors to the ISDN or POTS interfaces on the rear panel of the unit.

Connect audio devices like microphones on XLR 1,2,3 and headphones on jack 6 35 on left and right panels. Eventually connect a monitor on the left XLR output connector.

Switch on your Scoopy+ pressing the on/off key  on the front panel and hold it for few seconds.

Select the network the appropriate network via the menu: Config ⇨ Network

Establish the link

- For a direct call, dial up the remote telephone number on the keypad
- Press  to make a call
- To display the last 5 called numbers, press  *
- To call a registered number, enter a letter. *

* for this 2 modes, select the profile via the joystick

To make the call, press again 

If the remote is busy or in case of bad connection, press . Then press twice on  to restart the call of last called number.

The status of the link is displayed when connection is established.

To release the Link, press 

2. Introduction

2.1. Fonctions

Scoopy+ is designed to enable radio broadcasters to conduct high quality live remote broadcasts, or two way commentaries with return cue, via various networks (depending on the model of the unit) :

- ISDN
- POTS
- IP Protocol network
- GSM 3G+ [1] network

In ISDN mode, Scoopy+ has the 5A System feature; on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

Scoopy+ includes an Ethernet Interface Module (EIM). With this module, the codec features a 100BaseT / 10BaseT, and the audio transmission can take place over an IP network through this interface..

Scoopy+ uses the SIP protocol, which eases the setting up of a Link. The operation is similar to setting a call over the ISDN or the POTS. The transmission can be done in two modes:

- Direct « peer to peer » transmission between two compatible units
- Use of a SIP server for the call setup

2.1.1. Coding algorithms

Scoopy+ contains an audio compressor/de-compressor that performs all necessary ISDN and POTS algorithms.

In ISDN mode, the user can select one of four operational audio standards:

1. **Phone mode** (G.711, 3,5 kHz)
2. **Live speech** (G.722, 7 kHz, low delay)
3. **Music CD quality** (MPEG Layer II,AAC-LC, 20kHz) [2]
4. **Live Concert** (MICDA 4SB, 15 kHz, proprietary low delay) [3]

In POTS mode, the user only has live speech mode (CELP, 7kHz).

In GSM mode, the user has only GSM codec mode (300 Hz – 3.5 kHz).

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[1] GSM Edge/3G+, afterwards called GSM

[2] On ISDN 20 kHz

[3] On ISDN 20 kHz

In GSM data mode, Scoopy + has the same algorithms than in IP mode.

In IP mode, the following algorithms are available:

1. **Phone mode** (G.711, 3,5 kHz)
2. **High Quality speech** (G.722, 7 kHz, low delay)
3. **High quality speech** (CELP, 7 kHz, low network bandwidth consumption)
4. **Highest quality** (MPEG Layer II,AAC-LC, 20kHz)

2.1.2. Audio inputs/ outputs

Scoopy+ contains an audio mixer, that enables three microphones to be mixed. The three mic inputs accept line level via adjustable PAD, when the source is either a recorder of a mixing console.

Two headphones sets can simultaneously be connected to the unit. Each headphone has its own volume adjustment. The outputs can be used to listen the received mix or sent signals.

To be noticed : the mix in headphone 2 can be sent on the line output.

The stereo line output can be connected to a preamplifier or another audio device.

2.1.3. Transmission

Using an ISDN line enables a transmission bit-rate of 64 Kbit/s or 128 Kbit/s.

Using a POTS line, Scoopy+ transfers data with a minimum rate of 12 Kbit/s and up to 24 Kbit/s.

Scoopy+ can work in many countries using various ISDN standards.

As ISDN protocol may vary from country to country, consult your AETA AUDIO SYSTEMS dealer before carrying your Scoopy+ abroad.

2.2. Applications

News report.
Live sport commentaries with local contributors.
Remote two-way interviews.
Remote contributions into studio discussions.
Live music concerts.

3. Setting up the Scoopy+

3.1. Power

The unit can be powered by 6 type « C » or LR14 alkaline cells. Heavy-duty alkaline cells or rechargeable NiMh cells can be used.
Caution: **OBSERVE THE RIGHT POLARITY WHEN INSERTING THE BATTERIES.**.

Only NiMh cells can be used due to the integrated charger in the unit. Please only consider alkaline cells as an emergency set. An automatic detection will avoid in such case the batteries to be charged.

IT IS STRONGLY RECOMMENDED THAT YOU DON'T USE LOW QUALITY, SALT OR ALKALINE CELLS. SUCH BATTERIES MAY CAUSE LEAKS AND DAMAGE THE UNIT.

The autonomy depends on the selected algorithm and the network. With fully charged batteries, we have from 3 to 4 hours autonomy. A battery-like indicator displayed on the screen indicates the residual level of batteries.

Given that it is not usually possible to know how far a given set of batteries has been discharged before use, ensure that you recharge them after each broadcast.

3.1.1. External DC supply

Scoop+ will also work on any external 8 to 15-Volts DC source. A typical source will be a car cigarette adapter. Connect your DC power cord to the connector on rear panel of the unit (labelled DC 8-15V 2A), and connect the other end into your DC power source.

Connector: Jack 3.5: Centre = +, Circle = Ground.

This accessory is available in our price list .

Warning: **Polarities must be strictly observed to prevent damage to the unit!**

3.2. Connection to the ISDN

Connect the RJ45 connector of the ISDN cable into the socket marked "ISDN" on the rear panel and plug the other end of the cable into the ISDN wall socket.

The ISDN modem of the Scoopy+ is an S/T.
You can select the correct ISDN protocol for a given country from the menu.

Given the various kinds of ISDN protocols used in different countries or inside PBXs, ISDN compatibility problems may occur. Please be sure to select the right protocol for the country you are in. In case of difficulty please contact your AETA dealer for advice

3.3. Connection to the POTS

Connect the RJ11 connector of the telephone cable into the socket on the rear panel marked "ANALOG", and connect the other end of the cable into the telephone wall socket.

The Scoopy+'s RJ11 socket will accept a 4 or 6 conductor modular plug, but only the 2 center conductors, (typically Red & Green) are used.

Caution: Every country has its own style of telephone connector. Consult your engineers, your local AAS dealer for further advice.

Dialing methods

Telephones dial numbers either by pulsing the line, (you will hear a "clicking" sound similar to that heard when dialing from a rotary dial telephone) or by sending audio tones (DTMF) Scoopy+ can dial using either pulse or DTMF tones.

Caution:

Please to not activate the call line signal as you may loose the link while transmitting

PBX and PABX applications

AETA AUDIO S.A can not guarantee that Scoopy+ will operate correctly under all possible conditions of connections to compatible PBXs. Any cases of difficulty should be referred in the first instance to AETA AUDIO S.A.

3.4. Connection to the GSM

In SCOOPY+ version with GSM module, the use on mobile networks strictly obeys same rules as with mobile telephone. On one hand you need a suitable SIM card for the considered network. You must insert the SIM card in the GSM module integrated in Scoopy+. On the other hand, Scoopy+ with its integrated module and antenna is autonomous, thus you don't need to use a mobile telephone.

To insert the SIM card inside Scoopy+, just plug it into the holder located on the rear panel of the unit.

Use a pen, press the left part of the holder to remove it. Insert the SIM card and slide the support deep in the hole.

If necessary, plug the antenna cable on the antenna connector on the rear panel. Scoopy+ also integrates an internal antenna, by doing so you have an automatic switching when connecting it. The GSM module is ready to operate.

If a PIN code has been entered in the SIM card, Scoopy+ will ask to enter this PIN code again when activating the GSM mode.

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4. Connection to IP: Initial setup

Before going further, connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the Scoopy+ automatically switches to the right 10 Mbit/s or 100 Mbit/s mode.
- “Straightforward” patch cables should be used for a connection to a hub or a switch. Conversely, a “crossed” cable might be needed for special configurations (e.g. a test connection to a PC).

4.1.1. IP Configuration (Ethernet Interface)

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network.

4.1.2. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select “DHCP” in the menu (MENU / SETUP / Net / Param / Network Config). The unit will then automatically find the DHCP server and automatically set the parameters. You can read the IP address (allocated to the unit by the DHCP server) in the “About” menu (MENU / TOOLS / Maintenance / About).

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the Scoopy+ to another network, as long as it is still connected to a DHCP server

4.1.3. “Static” IP configuration

When there is no DHCP server, you have to enter the settings manually, using the menu (MENU / SETUP / Net / Param / Network Config / MANUAL / etc.). The IP address must be “free”, i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed. The following has to be entered:

Parameter	Notes
IP address	Must be unique on the network
Network mask	A typical value is 255.255.255.0
IP Gateway	Default gateway
DNS	Domain Name Server – this information can be avoided

All parameters are in the form nnn.ppp.qqq.rrr (see au-dessous 6.2 page 23 how to enter the “.” key).

☞ *Note: in contrast to the configuration with DHCP, the “static” setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

4.1.4. Checking the IP configuration

The above configuration is kept in the unit's memory, and reloaded at each start. It is recommended to restart the unit right after the initial setting, to ensure that everything is OK.

To check the setting, you can read the IP address in the "About" menu (MENU / TOOLS / Maintenance / About).

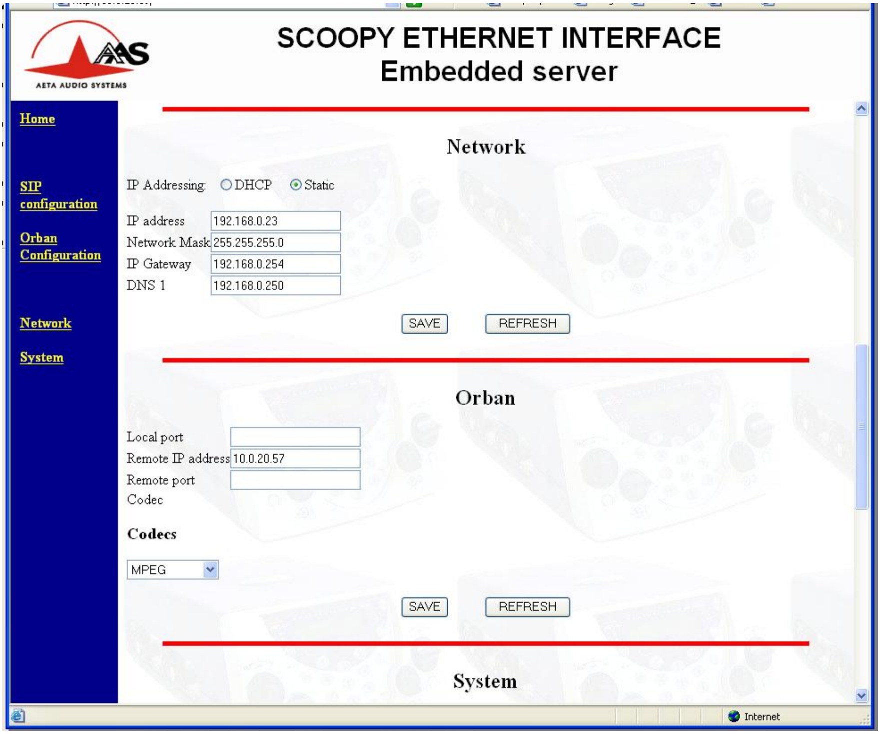
You can then also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) "ping *ipaddr*", where *ipaddr* is the IP address of the Scoopy+

If the response is positive, then you can proceed with the rest

4.2. Use of the embedded html server

From a computer connected to the same network, open an html browser window and enter the IP address of the Scoopy+ in the "address" or "URL" field. This gives access to the html server that is embedded in Scoopy+.

A typical screen copy can be found hereunder.



If you click “Network” on the left, you can get a display similar to the above. It is possible to change settings, and click the “SAVE” button[1] to write them into Scoopy+. “REFRESH” reloads the page from the unit to update the display.

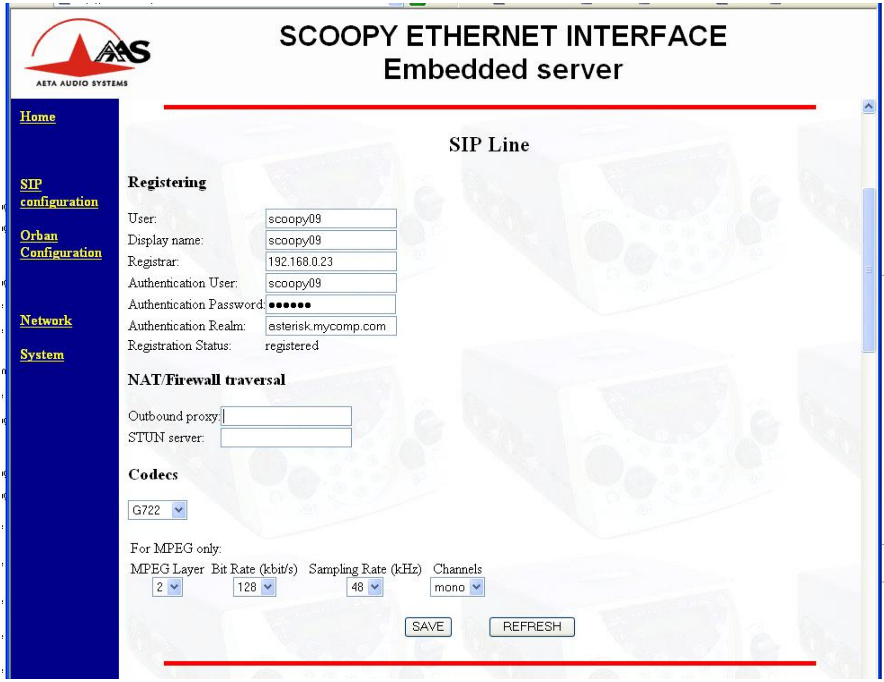
The network settings can be updated from this page, **but**:

- Obviously it is not usually possible to do the initial setting in this way!
- Be careful before changing these settings, as a wrong setting here can make you loose control over the unit… (In such event, go back 3.5.1 above)

[1] Important notice : the SAVE button only uploads a section (enclosed between two bold horizontal lines), unlike the REFRESH button, which refreshes the whole page.

4.3. SIP registration and configuration data

Scoopy+ uses the SIP protocol to establish links with compatible units. This protocol is a standard; you will find a quick presentation in annex . If you use a SIP proxy server, you need to configure your unit and save its configuration. The following is an example screen copy, and some comments about the displayed data



Item	Notes
User, Display name, Authentication user	Refer to the network administrator and/or the administrator of the SIP server; Often these three parameters have the same value, as here, but they may be different.
Authentication password	Refer to the network administrator and/or the administrator of the SIP server
Registrar	IP address of the SIP registrar; a symbolic name (e.g. sipsrv.mycomp.com) is accepted, if recognised by the DNS. *Can be also read from the menu* (MENU / TOOLS / Maintenance / About)
Authentication realm	Enter here the "realm" value of the SIP server. If it is not known, simply enter "realm".
Registration status	(read only data) Shows that the unit is (or is not) successfully registered on the server. *Can be also read from the menu* (MENU / TOOLS / Maintenance / About)
Outbound proxy	An outbound proxy is one way of getting access through a NAT router or a firewall; Refer to the network administrator and/or the administrator of the SIP server for this setting
STUN server	A STUN server is also one means of getting access through a NAT router. If such server is available, enter here its IP address or domain name.
"Codecs" sub-section	This defines the desired encoding for outgoing calls *(The unit adjusts automatically for incoming calls)* This setting can also be done from the keypad

Make sure to click the "SAVE" button located at the bottom of this section if you want to actually write your changes into Scoopy+.

Some special settings are reserved at the moment. Do not change these parameters from their initial settings: "Codec mode" (leave SIP selected), Orban (do not save any change). Be careful with the security password. This optional feature is left blank in the initial factory setting.

The registration data do not have to be changed often in normal operation. In fact, they may be still valid even after the unit moves to another location, even though its IP configuration changes.

5. SCOOPY+ structure

5.1. Front view

Figure 1 – Front panel

1 – Headphone 1 : Local/Cue (external ring) and volume adjustment (central knob)
2 – Input 1 potentiometer

3 - “red phone” : hang up

4 – OLED Screen

5 – Input 1 : mute mic activation
6 – keypad

7 – Input 2 potentiometer

8 Headphone 2 : Local / external ring) and volume adjustment (central potentiometer)

9 - “green phone” : call key

10 – Joystick

11 – Switch ON/OFF

5.2. Rear view

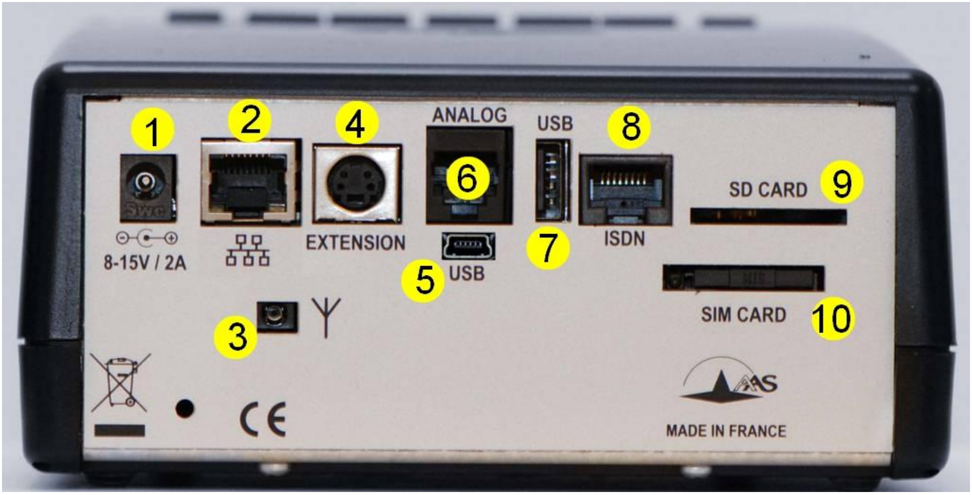


Figure 2 – Rear panel

1. External DC Jack
2. Ethernet RJ45 Jack
3. External antenna connector for Integrated GSM module
4. Audio device extension connector, compatible with Mixy
5. USB (Mini-B) to connect to PC
6. POTS RJ11 Jack
7. USB interface to connect with any USB peripheral
8. ISDN RJ45 Jack
9. SD card holder
10. removable holder for SIM card

5.3. Scoopy+ status

There are LED’s on the front panel providing the following information:

- Info (2 yellow LED’s): remote loops status.
- Dec: When « green »: indicates that a successful connection exists and the Scoopy+ is decoding the received signal from the network. When “ red”, it indicates a network problem no audio synchronization
- DC/Charge : When « green » Scoopy+ uses an external DC source. When « red » it indicates that batteries are under charging process.

- ON (red) : when « on » it indicates that the X input is enable (open)
- LIM (green/ red): When green “on”, the limiter or the compressor is on stand by on input X. When red “on”, the limiter or the compressor is functioning.

5.4. General synoptic diagram

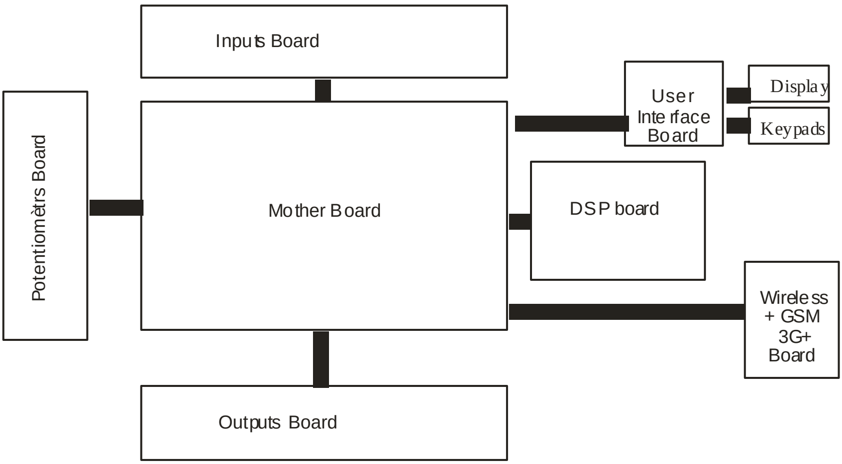


Figure 3 – Scoopy+ synoptic

6. Audio section

6.1. Encoding and decoding

Scoopy+ includes a wide range of coding algorithms. First, one can select among algorithms compliant with ISO and ITU-T[1] recommendations:

- G711;
- ITU-T G722 (mono at 64 Kbit/s);
- MPEG Audio Layer II at 48, 32, 24 or 16 kHz, with programmable channel mode and bit rate ;
- AAC-LC 48, 32, 24, 16 kHz with adjustable rates (option)

MPEG Audio and G722 algorithms comply with ITU-T J52 recommendation for ISDN transmission.

Besides, other algorithms are available, that are so-called "proprietary" because they do not comply with enforced standards:

- Proprietary MPEG Layer II at 64 Kbit/s or 128 Kbit/s (for compatibility with ISDN codecs that are not compliant with the J52 recommendation) ;
- 4SB ADPCM, running in mono at a 128 Kbit/s rate; the bandwidth with this algorithm is 15 kHz ;
- TDAC mono, running at 64 Kbit/s, with a 15 kHz bandwidth; available as an option.

☞ *importante Note:Availability of algorithms depends of existing networks (POTS, ISDN, IP, GSM) as well as the version of the product..*

6.1.1. Notes about G711

G711 is the standard coding used for voice transmission on public telephone networks. This algorithm is used for links (via ISDN) with telephones or hybrid devices.

6.1.2. Notes about G722

With G722 coding, three synchronization modes are available:

- "Statistical recovery" byte synchronization method (alias SRT) ;
- H221 synchronization; in this case, 1.6 Kbit/s from the compressed data are used for this.
- H221 synchronization and H242 protocol.

[1] former CCITT

H221 synchronization is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronization but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

6.1.3. Notes about J52 and MPEG coding

The ITU-T J52 recommendation was defined in order to allow the interoperability of various equipments over the ISDN[1], using common coding standards. It includes the following features:

- Interoperation procedures as per ITU-T H242 recommendation ;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes).

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.
As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

[1] J52 is not needed nor applicable to leased line connections

6.1.4. Notes about TDAC

As an option, the codec can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation; this is a transform coding based on an MDCT (Modified Discrete Cosine Transform), encoding a 15 kHz bandwidth mono signal at a 64 Kbit/s bit rate. When the option is installed, three modes are available:

- TDAC mono full-duplex, running at 64 Kbit/s, with a 15 kHz bandwidth ;
- G722/TDAC : G722 encoding, TDAC decoding, running both in mono at 64 Kbit/s ;
- TDAC/G722: TDAC encoding, G722 decoding (with SRT), running both in mono at 64 Kbit/s; this mode is symmetric to the previous one.

6.1.5. Symmetric or asymmetric codec modes

The codec allows two communication modes:

Symmetric communication: in this mode, the encoder and decoder both use the same coding algorithm with the same configuration (channel mode, etc.). In this case, the communication is strictly symmetric full-duplex, with exactly the same coding configuration used in both directions (local to remote and remote to local). This is usually required when using proprietary algorithms.

Asymmetric communication: this mode is used for applications requiring different coding configurations in the two directions. The J52 protocol allows such mode. To give some examples, it is possible to transmit MPEG in one direction and G722 in the other one.

With the TDAC option, asymmetric modes are also available wherein one direction is G722 coded while the other one is TDAC coded. Such mode is useful e.g. in order to get a low delay return path encoded in G722 while the send path is encoded with higher quality but a higher delay.

6.2. Audio Interfaces

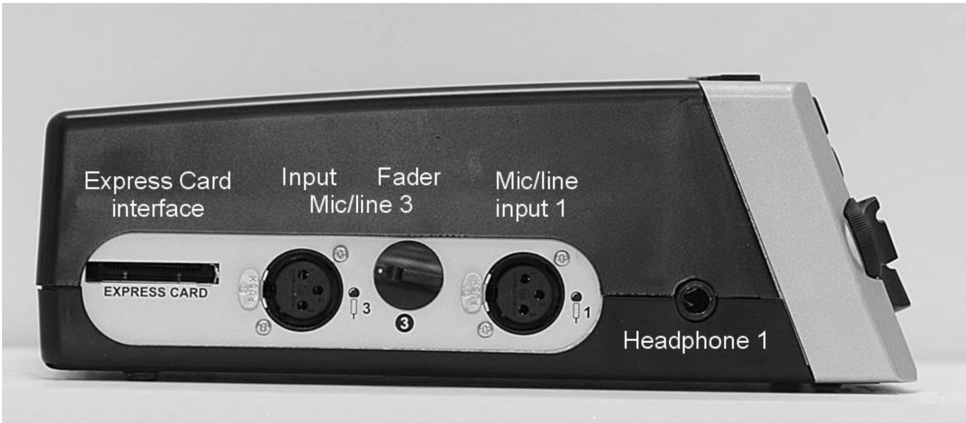


Figure 4 – Left panel

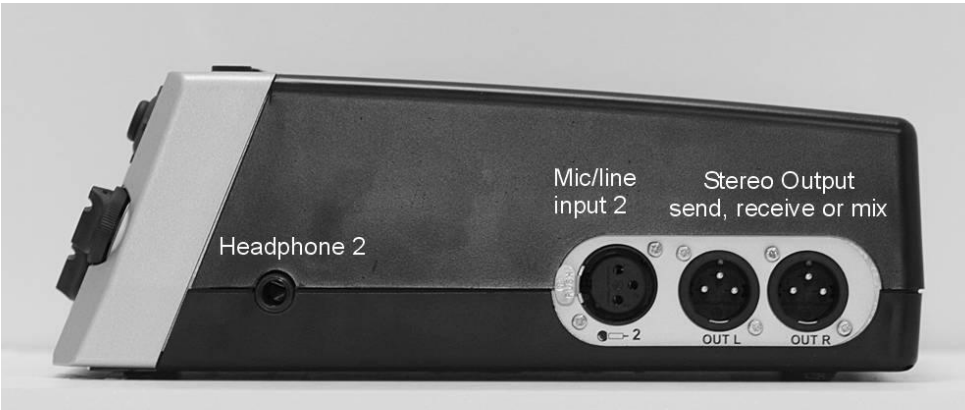


Figure 5 – Right panel

6.2.1. Inputs

The mixer features three mic/ line inputs with microphone powering.

The following elements are available for each mic/line input:

- Input connector: female XLR;
- Pad switch to adjust the gain on each input, located on the front panel of SCOOPY+ (fig.1- ref 2 & 3)
- A mute activation pad located on the front panel

By menu [1]

- Gain pad (to adjust by step of 16 dB from 0 to 48dB).
- Microphone power selection switches
  None: dynamic microphone or line live
  Phantom: phantom power supply (48V or 12V switch configuration)
  T12 : "Tonadder" (12V) for certain static microphones
- Each input has a high pass filter of second xxxx 50Hz.
- Each input has a limiter preset at -8dBFs.
- Each input can be routed on left or right channel of the codec.

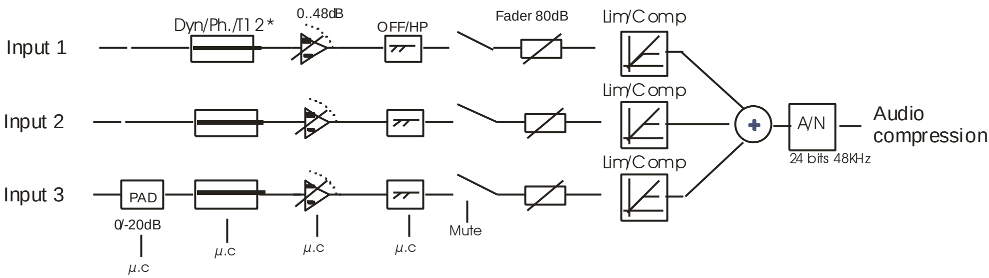


Figure 6 – Audio Inputs

[1] Voir menu Audio

Format	symmetrical
Connector	3-pin female XLR socket
Microphone powering	Phantom 48V or 12V, optional Tonadder 12V
Maximum input level	+19 dBu (+39dbu Channel 3)
Input stage sensitivity adjustment	+0 to +48 dB by steps of 16dB
Input impedance	10 kΩ
CMRR	>80dB @ 1kHz

Table 1 – Input interface

6.2.2. Outputs Interface

The mixed audio signal from inputs is available on the 2 headphone outputs and the line input.
Local audio from the inputs can be mixed with the return audio signal on each headphone. The return audio signal is present in the headphone mix via the Local / Cue Mix Balance potentiometer on the front. (See Page 11 – Front panel, figure 1).
By default, the return program audio is assigned to the output line. You can assign local audio program or the headphone 2 mixed audio signals to the output by menu[1].

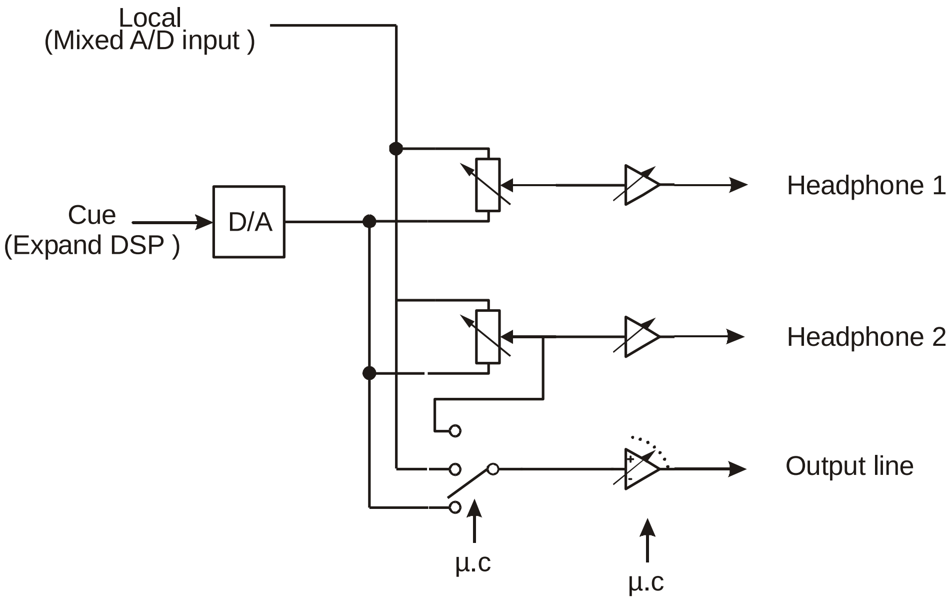
Line Out Interface:

Format	Symmetrical
Connector	3-pin male XLR socket
Maximum Output level	-11 to +22dBu (adjustable by menu)
Output impedance	$\leq$ 50 Ω
Output symmetry	> 40 dB

Interfaces casques :

Connecteur	6.35mm jack socket
Maximum output level	+20 dBu
Output impedance	$\geq$ 16 Ω

[1] See audio menu section Page **Erreur ! Signet non défini.**



µ.c : micropressor command

Figure 7 – mono or stereo audio outputs

6.3. Audio Performances

A) Analog performance

Conditions de mesure :

Measurement condition:

- AD/DA Loop
- Sampling frequency: 48kHz

Maximum Gain (Input to Output)	+87 dB
Signal to Noise ratio	84 dBrms
Bandwidth	20Hz – 20 000 Hz ± 0,5 dB
Distortion (THD+N)	< 74 dB (0,02 %) à 950 Hz

Table 2 –Audio performances

B) In ISDN mode

Data rate	Sampling frequency	Bandwidth	Delay	Algorithm
128 kbit/s	48kHz	20Hz - 20kHz	137 ms	MPEG Layer II
128 kbit/s	32kHz	20Hz – 15kHz	202 ms	MPEG Layer II
128 kbit/s	24kHz	20Hz – 10.4 kHz	268 ms	MPEG Layer II
128 kbit/s	16kHz	20Hz – 7.2 kHz	398 ms	MPEG Layer II
128 kbit/s	32kHz	20Hz – 15 kHz	7 ms	4SB ADPCM
64 kbit/s	48kHz	20Hz – 20kHz	163 ms	MPEG Layer II
64 kbit/s	32kHz	20Hz – 13.4kHz	202 ms	MPEG Layer II
64 kbit/s	24kHz	20Hz – 10.4 kHz	268 ms	MPEG Layer II
64 kbit/s	16kHz	20Hz – 7.2 kHz	400 ms	MPEG Layer II
64 kbit/s	32kHZ	20Hz – 15kHz	80 ms	TDAC
64 kbit/s	16kHz	20Hz - 7kHz	11 ms	G722 SRT/H242
64 kbit/s	16kHz	300Hz – 3.5kHz	17 ms	G711- phone

Table 3 – ISDN Mode

Note: In MPEG Layer II without J52, Scoopy is compatible with other manufacturer codecs.

C) POTS mode - CELP Algorithm

Data rate	Audio quality
12 kbit/s	3,6kHz
14,4 kbit/s	4,3kHz
16,8 kbit/s	5,1 kHz
19,2 kbit/s	5,7 kHz
21,6 kbit/s	6,3 kHz
24 kbit/s	**7,2 kHz**

Bandwidth : 40 Hz to 7 kHz (@ 24 kbps data rate)

24 kbit/s can typically be achieved in all countries that support V.34 modems on their public switched networks.

Table 4 – CELP

6.4. Audio monitoring

The levels on the bar graph on the front panel of the Scoopy+ indicate the peak level of the mixed audio signal**.** The level displayed is registered at the analog-digital converter overloading level. The ‘reference’ level can be fixed by the audio menu.

6.5. International Sound

Input 3 can be set by menu as an international sound input. Then this mode is on, the input 3 is sending on the right headphone in place the return signal.

6.6. Loops activation

When this function is activated, the codec transmits to the remote unit the status of two isolated current loops. The remote unit then opens or closes relay contacts according to the transmitted status. Conversely, as the function is bi-directional, the codec activates its two relays (“dry” isolated contacts) depending on the status of the two current loops on the remote unit.

MENU ⇨ TOOLS ⇨ MISCELLANEOUS ⇨ Aux. Fonctions ⇨ Relay (keypad)

F1 and F2 keys are used when the connection is established.

When the two remote loops are activated, status of the loops can be seen via the INFO diodes on the front panel.

A typical application is the transmission of an “on air” signal; the contact closure may be used for e.g. switching on a lamp or starting other devices.

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7. Scoopy+ functionning

7.1. Introduction

The audio mixed signal for the input is converted to digital by the AD converter and processed by the Scoopy+ mixer. Then data are transmitted via the internal synchronous POTS or ISDN modem to the POTS networks to another remote Scoopy+, or any other ISDN compatible codec or via a wired or wireless IP network.

Based on a very powerful DSP, the codec uses an algorithm to compress the digital audio signal so to reduce the quantity of digital audio data.
At the other end of the link, the audio codec decompresses the original audio signal with low losses or interferences in an extremely low delay.

7.2. User interface

The user interface consists of a lexan matrix keypad and a LCD display. The keypad has three sections:

- The first section is a 4x3 matrix including the numbers from 0 to 9, "*", "#".
  Some keys have many functions:
  2, 3, 4, 5, 6, 7, 8, 9, and 0: for accessing to letters display on the key, press the key several times.
  Note: Space character is available on the "1" key.

- The second section is a joystick to access the menus.

- The third section is the special Keypad functions.
  OK Key to validate a choice.
  ESC Key to escape from a menu
  "Green phone", key to make a call.
  "Red phone" key to on hook a call.

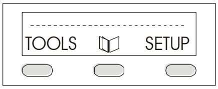
  Mute (Mic 1, 2, 3): Key to press to activate or close the audio input.

Note: To enter specific characters

You can change at any time the way to enter them by pressing the « # » key. By pressing repeatedly on this key you access other characters.

7.3. Scoopy+ menus

Main menu.



To scroll in the menus and sub-menus use the function keys which actions are mentioned on the screen.

At any time you can return to the main menu by pressing the ESC key.

Note: the symbol between "TOOLS" and "SETUP" means: "DIRECTORY"

If you press "green phone" key, you access to the 5 last called numbers.
If you enter a letter, you access to the remote directory.
If you enter a number, you can make a direct call.

Note: If you have a restricted menu, you need to enter your password as per in the menu/tools/maintenance/password to access the menus.

7.3.1. Relay control

When a link is established, and auxiliary functions are activated, a menu « rel » enables to change the status of this information. On the screen, local loops status are displayed, INFO LED's give remote loops status.

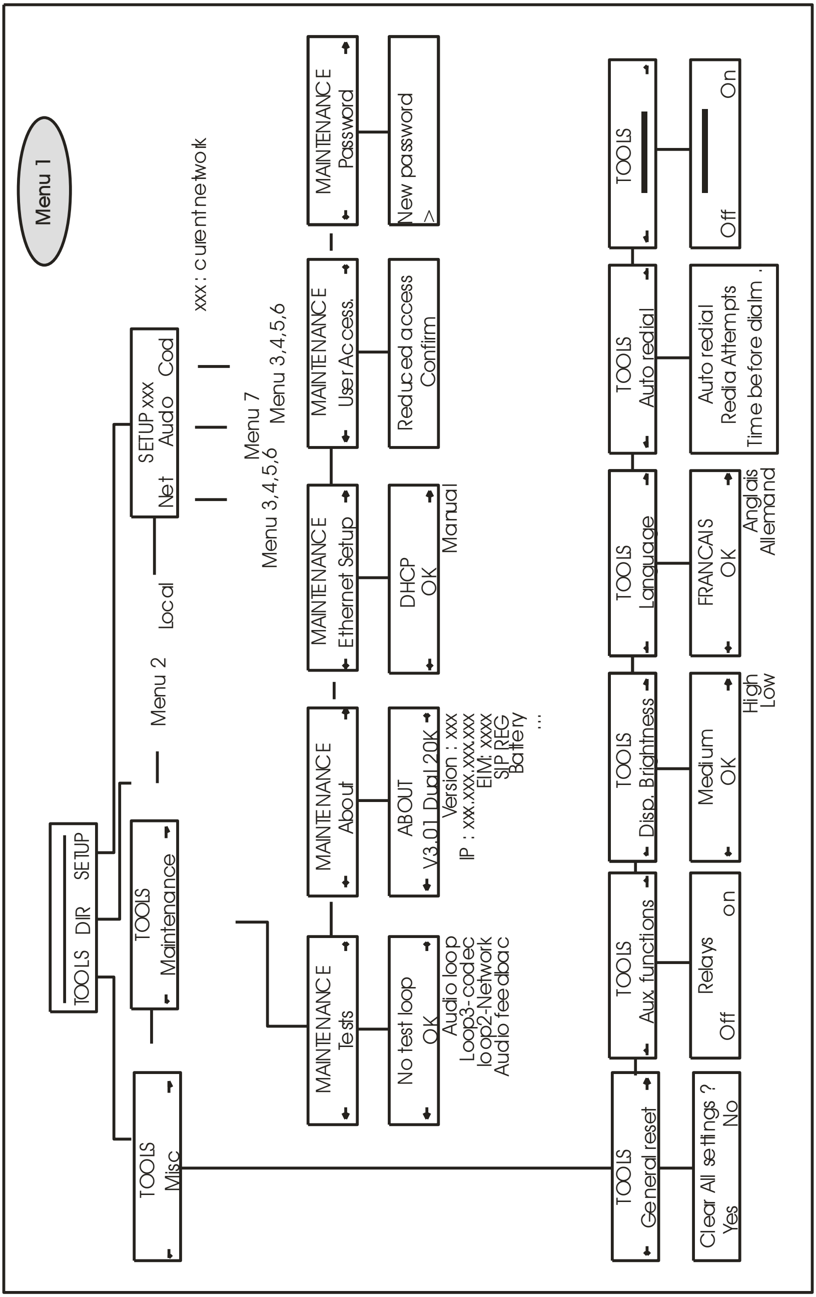
7.3.2. Mute monitoring

Au repos et en communication, l'utilisateur a la possibilité d'activé ou non les entrées audio par l'intermédiaire des touches de fonction..

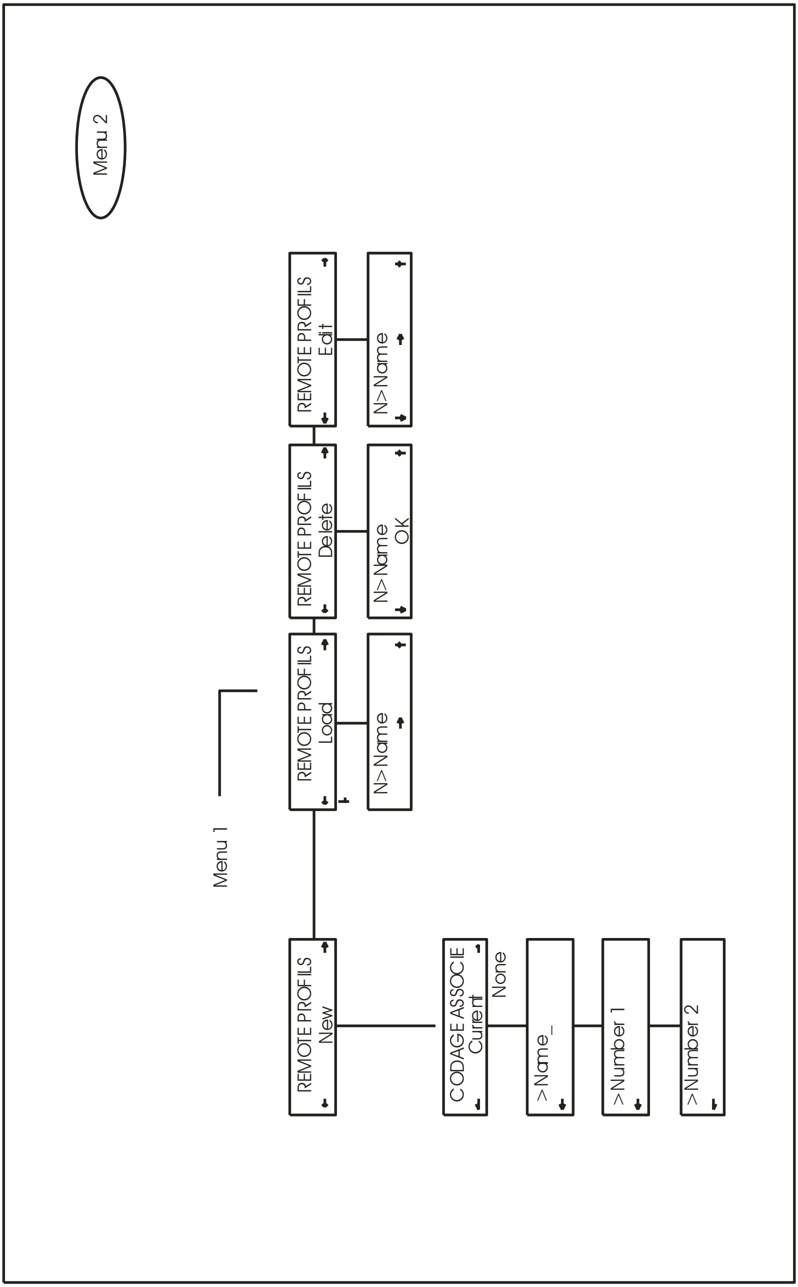
7.3.3. Scoopy+ default configuration

The Scoopy's "General Reset" (set default configuration) is useful to configure the modem in case communication difficulties are encountered or if you think that wrong network parameters are set up

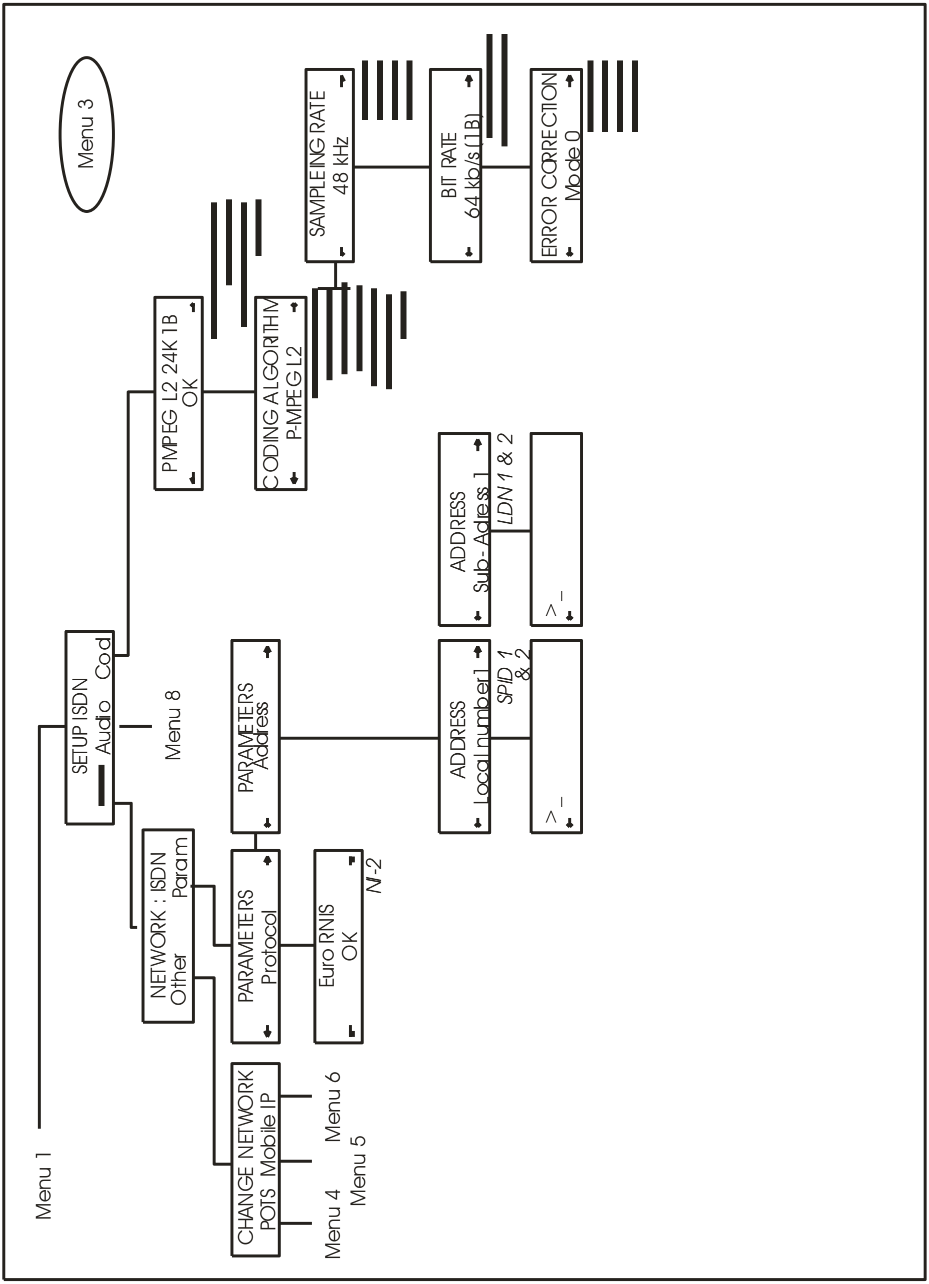
Note : when you make a software update on your Scoopy+ (download a new software version), please make sure to do a General reset

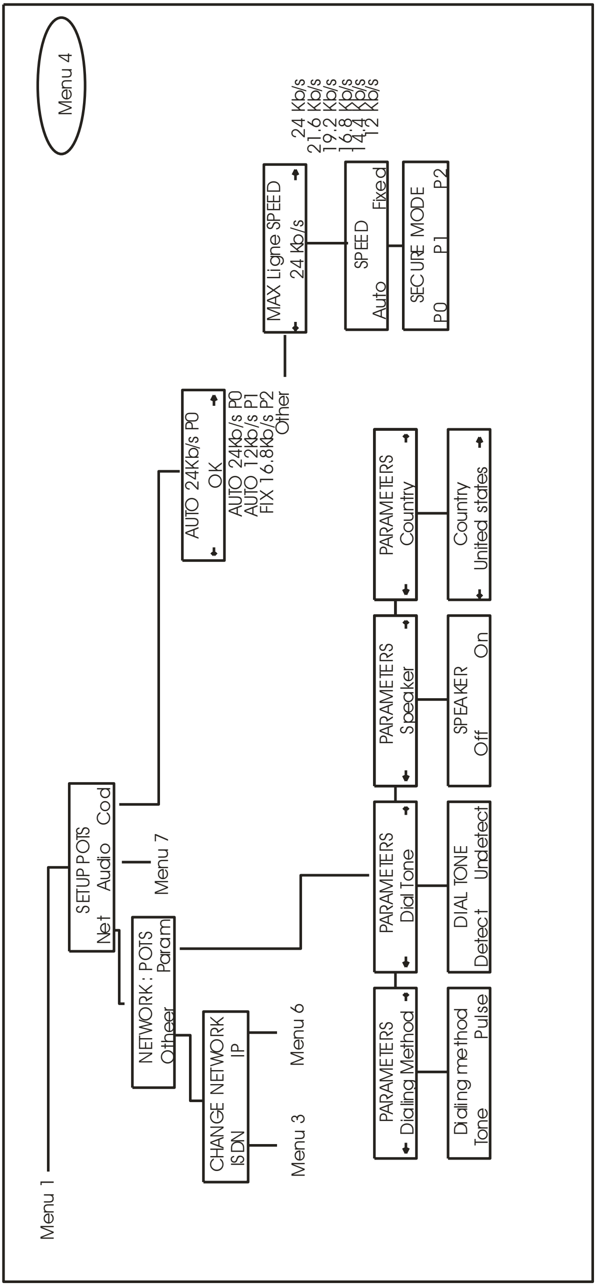

Menu 1
TOOLS DIR SETUP
TOOLS
Misc
TOOLS
Maintenance
Menu 2
local
SETUP xxx
Net Audio Cod
xxx : curent network
Menu 3,4,5,6
Menu 7
Menu 3,4,5,6
MAINTENANCE
Tests
No test loop
OK
Audio loop
Loop3-codec
loop2-Network
Audio feedbac
MAINTENANCE
About
ABOUT
V3.01 Dual 20K
Version : xxx
IP : xxx.xxx.xxx.xxx
EIM: xxxx
SIP REG
Battery
...
MAINTENANCE
Ethernet Setup
DHCP
OK
Manual
MAINTENANCE
User Access.
Reduced access
Confirm
MAINTENANCE
Password
New password
>
TOOLS
General reset
Clear All settings ?
Yes No
TOOLS
Aux. functions
Relays
Off on
TOOLS
Disp. Brightness
Medium
OK
High
Low
TOOLS
Language
FRANCAIS
OK
Anglais
Allemand
TOOLS
Auto redial
Auto redial
Redia Attempts
Time before dialm .
TOOLS
Off On

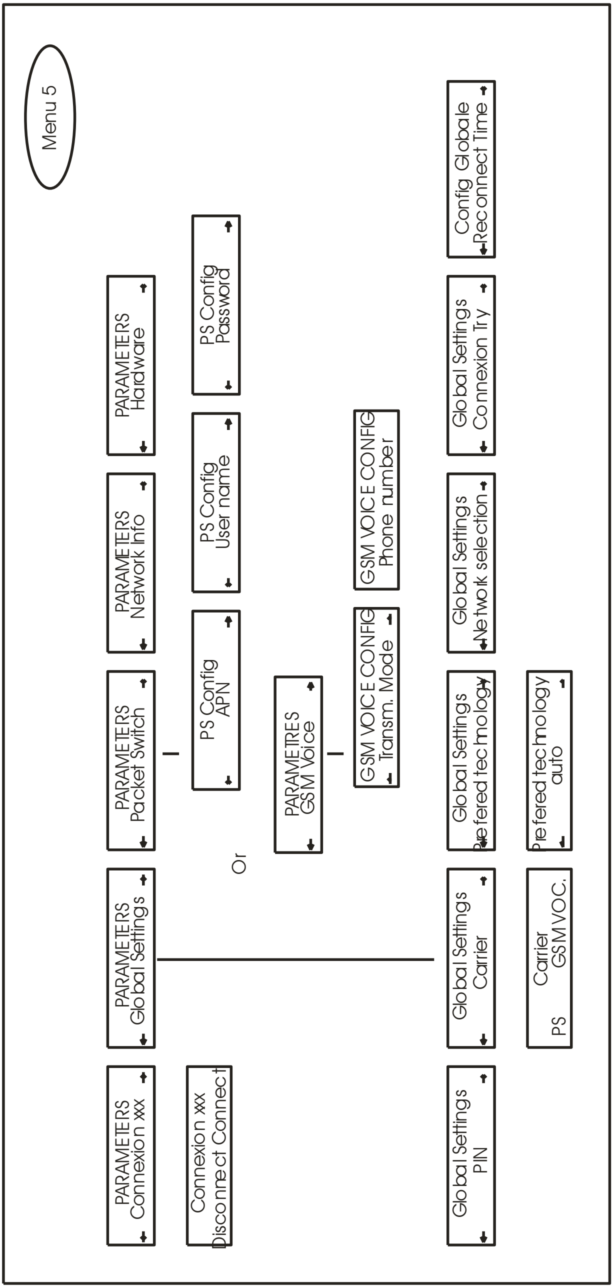
AETA AUDIO SYSTEMS

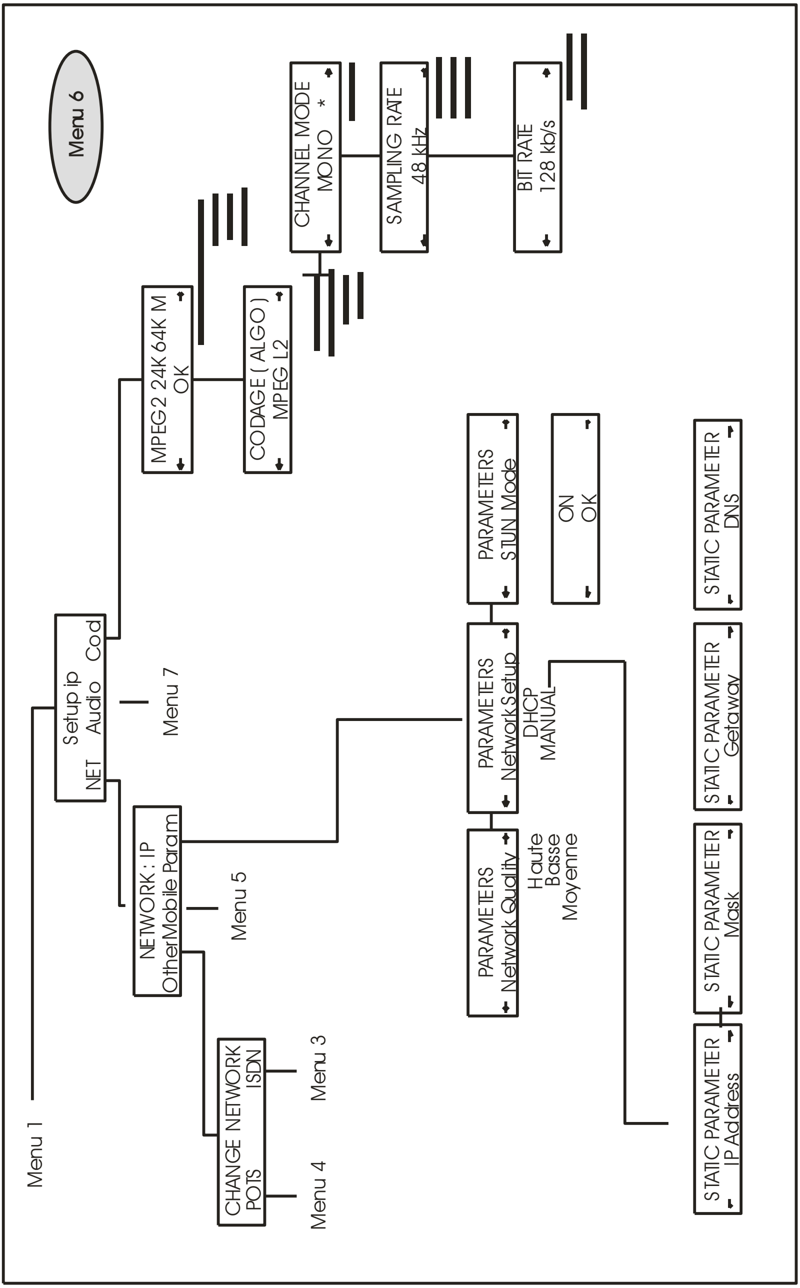


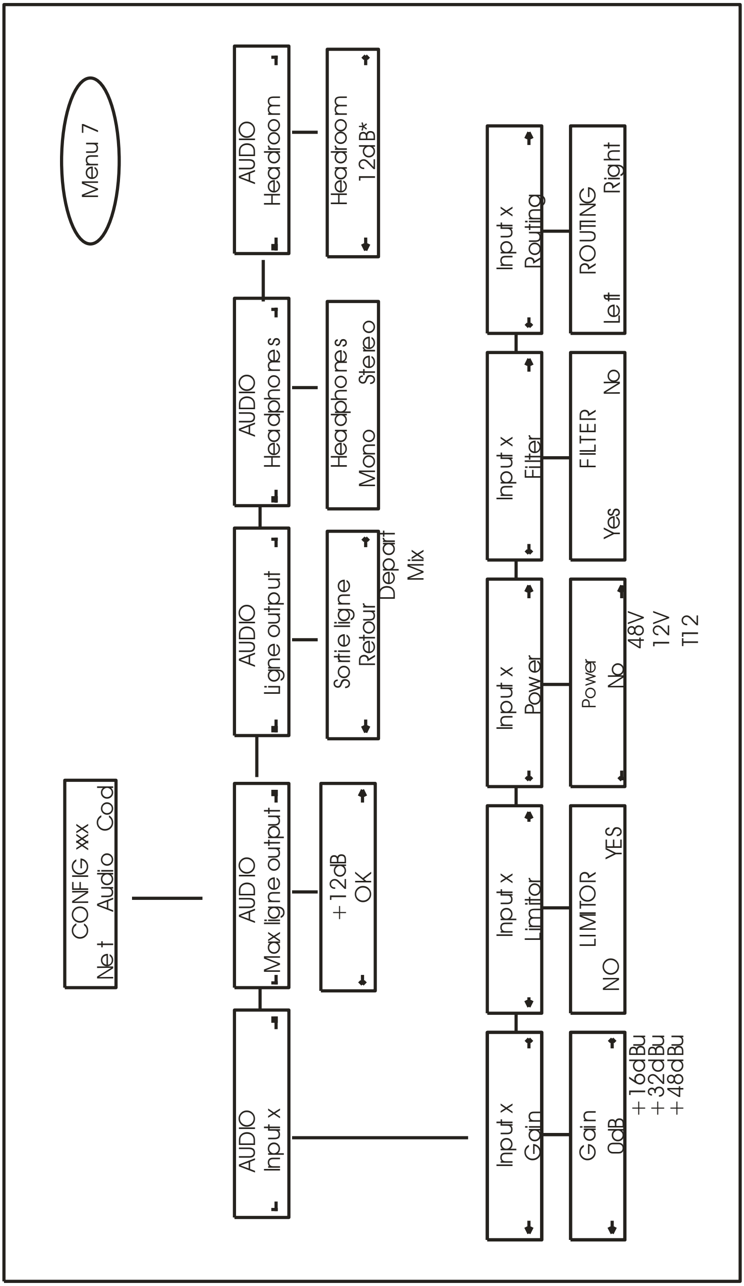
AETA AUDIO SYSTEMS




Menu 4
MAX Ligne SPEED
24 Kb/s
24 Kb/s
21.6 Kb/s
19.2 Kb/s
16.8 Kb/s
14.4 Kb/s
12 Kb/s
SPEED
Auto
Fixed
SECURE MODE
P0
P1
P2
AUTO 24Kb/s P0
OK
AUTO 24Kb/s P0
AUTO 12Kb/s P1
FIX 16.8Kb/s P2
Other
SETUP POTS
Net
Audio
Cod
Menu 7
NETWORK : POTS
Other
Param
CHANGE NETWORK
ISDN
IP
Menu 6
Menu 3
Menu 1
PARAMETERS
Dialing Method
Dialing method
Tone
Pulse
PARAMETERS
Dial Tone
DIAL TONE
Detect
Undetect
PARAMETERS
Speaker
SPEAKER
Off
On
PARAMETERS
Country
Country
United states


Menu 5
PARAMETERS
Connexion xx
PARAMETERS
Global Settings
PARAMETERS
Packet Switch
PARAMETERS
Network Info
PARAMETERS
Hardware
Connexion xx
Disconnect Connect
Or
PS Config
APN
PS Config
User name
PS Config
Password
PARAMETRES
GSM Voice
GSM VOICE CONFIG
Transm. Mode
GSM VOICE CONFIG
Phone number
Global Settings
PIN
Global Settings
Carrier
Global Settings
Prefered technology
Global Settings
Network selection
Global Settings
Connexion Try
Config Globale
Reconnect Time
PS Carrier
GSM VOC.
Prefered technology
auto


Menu 6
Setup ip
NET Audio Cod
Menu 1
Menu 7
NETWORK : IP
OtherMobile Param
Menu 5
CHANGE NETWORK
POTS ISDN
Menu 3
Menu 4
MPEG2 24K 64K M
OK
CODAGE (ALGO)
MPEG L2
CHANNEL MODE *
MONO
SAMPLING RATE
48 kHz
BIT RATE
128 kb/s
PARAMETERS
STUN Mode
ON
OK
PARAMETERS
Network Setup
DHCP
MANUAL
PARAMETERS
Network Quality
Haute
Basse
Moyenne
STATIC PARAMETER
IP Address
STATIC PARAMETER
Mask
STATIC PARAMETER
Getaway
STATIC PARAMETER
DNS


Menu 7
CONFIG xxx
Net Audio Cod
AUDIO
Input x
AUDIO
Max ligne output
+12dB
OK
AUDIO
ligne output
Sortie ligne
Retour
Depart
Mix
AUDIO
Headphones
Headphones
Mono Stereo
AUDIO
Headroom
Headroom
12dB*
Input x
Gain
Gain
0dB
+16dBu
+32dBu
+48dBu
Input x
Limitor
LIMITOR
NO YES
Input x
Power
Power
No
48V
12V
T12
Input x
Filter
FILTER
Yes No
Input x
Routing
ROUTING
Left Right

8. How to Set-Up Profiles on Scoopy+

8.1. What is a profile?

A profile is a non-volatile, pre-programmed memory location stored within Scoopy+ which functions very similar to the auto dial memory locations on an average telephone.

A **remote profile** can contain an ISDN or POTS number with specific parameters associated with that number. The remote profile can contain the name of the location to be dialed and its specific algorithm. You can create up to 50 unique remote profiles on the Scoopy+. If the remote profile is an ISDN type, you can have two numbers stored (one number for each B channel).

A remote profile has similar functions than a stored number for a standard telephone ; it enables to establish very quickly a Link with the pre-registered correspondent.

A **local profile** (named local setup for making the difference) can contain ISDN or POTS local parameters. The local setup can contain the name and all network parameters like local address in ISDN. You can create up to 50 local setups on the Scoopy+.

A local profile enables you to find out the settings and parameters of the network access , associated to the location where you Scoopy+ is connected.

Note: In GSM mode, local profiles don't exist

8.2. How to manage profiles on Scoopy+?

You can access and register your profiles directly from the keypad and screen in Scoopy+'s front panel.

You can download p to 100 profiles.

NB : All remote profile can be a POTS profile or an ISDN profile.

8.2.1. How to manage remote Profiles?

From the Main Function Menu, select "DIRECTORY" symbol, and press the key under it. After having selected "REMOTE" choice, you can select different actions:

- "Load" for loading in memory a profile.
- "Delete" for deleting a profile.
- "Edit" for looking each elements of the profile.
- "New" for creating a new profile.

"Load"

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begin with this letter.

With the center extended display key, you scroll each profile items.

For calling with the displayed profile, you have just to press the "green phone" key.

For loading in memory the profiles, press the "OK" key.

“Delete”

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begins with this letter.

With the center extended display key, you delete the display profile

“Edit”

With the left and right extended display key, you can scroll the remote profiles. If you enter a letter on the keypad, the profile list scroll to the profile whose the name begins with this letter.

With the center extended display key, you scroll each profile items.

Press OK key to modify one or many items of the display profile.

“New”

At first you should select the network: Analog/POTS or ISDN.

After that, you have the choice to associate an algorithm (current algorithm configuration) or not to your profile. In this case, select “none” in “Associated Coding” .

If you don’t associate an algorithm to your profile, when you will make a call with this profile, scoopy+ will use the current algorithm configuration.

The next stage is to enter a name for your profile (press the suitable key as many times as necessary and press OK .to valid.

Now, Scoopy+ asks the user to enter one or two numbers (depending on algorithm configuration). If you don’t enter numbers, when you will make a call with this profile, Scoopy+ will ask you to enter the numbers.

8.2.2. How to manage local setups

From the Main Function Menu, select 🕮. After having selected “LOCAL” choice, you can select different actions:

- “Load” for loading in memory a local setup.
- “Save as” for creating a new local setup.
- “Delete” for deleting a local setup.

In a local setup memory we save the current network configuration. The “save as” function enables the user to register the current network parameters in a new profile.

For example: If you are in ISDN, we save the ISDN protocol, the local address and the local sub-address.

9. To make a Link over ISDN

Note: The following is valid for both POTS and ISDN mode.

Warning: In ISDN mode with some PBX's, you must enter your local number and your SPID number (for the USA) prior making a call.

9.1. Initiating a call

There are 3 ways to initiate a call:

- Dialing with a profile
- Direct Dialing
- Re-dialing the previous number.

9.1.1. Dialing Using a Profile Number

From the Main Function Menu, select . After having selected "REMOTE" choice, press the key "Load".
Note: You arrive directly in the remote profile list, when you enter a letter under the main menu.

Select the profile number OK and press to initiate the call.

"Call XXXXXXXX" appears on the screen and is dialed automatically.

9.1.2. Direct Dialing

Enter the telephone number and press .

If you call twice same number (In ISDN), you need to press again .

Note: In the case that you have 2 numbers, if you don't enter the second number, we call twice the first number.
A message to indicate that the call is in progress is displayed on the screen.

Note*:*
- *The number length is limited to 23 digits and may be displayed on 2 lines.*
- *Insert a "*" between number and sub-number in ISDN mode.*
- Insert a "*" for wait in POTS mode

9.1.3. Re-Dialing the Previous Number

From the main menu, press , the last called numbers appear on the screen; select the one needed with left and right key then press again . Actually, you are in a short list of the five last called numbers.

"Call in progress" along with the redialed number is displayed on the screen.

Note: We don't re-load configuration, we use the last configuration used (current now).

.

Note:

In case of mistake you may come back at the beginning of the menu by pressing .

Note:
As soon as the local and remote Scoopy+ are connected, the CONNECT result code is displayed.
If a connection cannot be established, the NO CARRIER result code will be displayed.
The bit rate is displayed in POTS mode.

9.2. Disconnecting a call

To end a call, press
"Wait..." is displayed, after awhile, Scoopy + is ready for the next call.
The main menu appears on the screen.

9.3. Receiving a call

As soon as the "Power on Initialization" phase is completed, Scoopy+ is ready to receive an ISDN call or a POTS call. You have just to adjust the levels in such a way that the audio level green and yellow LED's indicate a normal operating range.

When a call is received, Scoopy+ automatically recognizes ISDN or POTS and establish the connection. Adjust your headphone level and your local feedback with the local/return mix balance if needed.

Then Scoopy+ is ready for full duplex audio communication.

9.4. Entering local Numbers (in ISDN mode)

From the Main Function Menu, go to “SETUP”, “NET”,”PARAM”. Use the suitable key to scroll to "Address" screen.

You have two-address configuration with each sub-address. You have a specific address and sub-address for each B ISDN channel.

A series of AT commands will be displayed and automatically return you to the “Address” menu

***Note**: In many case, the sub address is not necessary*

9.5. Entering SPID Numbers (USA)

In the USA, some ISDN circuits require two SPID numbers and two LDN (Local directory number), one SPID for each B channel, in addition to the local dialing number. Scoopy+ can be manually programmed using the keypad.

From the Main Function Menu, go to “SETUP”, “NET”,”PARAM”. Use the suitable key to scroll "Address" screen.

Note:

- You should enter SPID 1 and LDN 1 local number at first, then SPID 2 , LDN 2
- Generally, the LDN is the 4 last digits of the SPID number.

10. POTS Information

10.1. Factory default configuration

The Scoopy+'s factory-set default configuration is suitable for most Scoopy+ transmission applications and are reloaded by the selected function:
“TOOLS”, “Misc”,”General reset”.

Your Scoopy+ is designed to operate over dial-up phone circuits with the following dialing and call monitor features:

- Multi-frequency signaling (Tone dialing method) or
  Loop-disconnect signaling (Pulse dialing method)

Remark:
The selection of the dialing method will be stored until the user has to modify his choice again even when the Scoopy+ is powered off.

- Operation in the absence of proceed operation (waiting for dial tone)
- Automatic answering
- Originating and answering handshake negotiations begin at the minimum rate specific to the maximum rate of the 2 equipments.
Automatic speed selection: Handshake negotiations fall back to a lower speed if necessary.
- Full dial progress detection (Dial tones detect).

Rem: This parameter must be "disabled" for calls originated from Switzerland and Italy.

Additional setting

- Fall back if negotiation fails at the highest speed (speed automatic)
- Maximum DCE Line speed = 24000 bps. (Default configuration)

10.2. *Network parameters*

10.2.1. Setting the optimal rate

The Scoopy+ with the lowest max line speed setting will determine the maximum connect rate.

a) When the speed mode function is set on "Automatic" adaptation (general reset Configuration) both modems will negotiate the highest transmission rate according to the quality of their current respective networks.

This rate is also limited at the lowest speed of the two max speed selected on the 2 units.

If the line quality is changing during the audio-transmission the modems will try to adapt consequently the data rate by fall back at a lower data rate and fall forward to the higher selected speed. During each re-negotiation the audio signal may be interrupted. If these "break down" appear, it is highly recommended to set the max line speed selection of one of the Scoopy+ at one level or two below the used connect rate.

b) When the speed function is set on "Fixed" at ONE of the both Scoopy+ unit, the 2 modems will be allowed to negotiate at only the lowest speed of the two max speeds selected.

They will neither "fall forward" nor "fall back".

If this select speed is too high for the possibility of one of the local network capacity, the modem will "NOT CONNECT" and a lower speed has to be selected by the user to obtain a solid connection at a reliable data rate. The user will not be able to modify the rate during transmission is on.

10.3. Error protection

This function reduces the short and occasional transmission errors causing glitches and dropout in the audio. Those errors can be founded particularly on long distance circuits and when connected to an in-house phone systems.

The audio quality could be slightly affected.

The user will only hear a short additional delay.

Three protection modes are available:

Protected 0 (Unprotected) is set by default (Factory Configuration)

- Is compatible with all Scoopy+ units.
- Keep the smallest transmission delay (coding: decoding) of 80 ms at 24Kbits.

Protected 1

Ought to be selected manually and recovers errors of 100 ms.

Protected 2

Ought to be selected manually and recovers errors of 250 ms.

Note: If errors still exist in protected mode 2, set the max line speed at the next lower speed.

Caution: **The same protection level configuration must be selected at the both end units.**

11. Setting up a Link over the Ethernet

A link is set up in a similar way as an ISDN or POTS link. The difference is mainly that instead of the telephone number, we use either an IP address, or a SIP URI (Uniform Resource Identifier).

11.1. Directly call an IP address

This is the most basic way of setting the link. It is suitable only if:

- The other unit is "directly" reachable, i.e. there is no NAT Router or firewall blocking the connectivity. The simplest case is when both units are on the same LAN.
- The IP address of the other unit is known.

To set the link, first set the desired encoding format (embedded server, or from the keypad MENU / SETUP / Cod).

Then enter the IP address and press

☞ *When operating in this way, it is preferable to leave blank the SIP registering data.*

11.2. Calling via a SIP server

This is the technique when both units are registered on a SIP server. In this case, each unit is identified by its SIP URI, in the form username@sipservername, like an email address. There is no need to know any IP address (and hence there is no problem if the IP address of a unit changes for whatever reason).

To set the link, first set the desired encoding format (embedded server, or simply from the keypad MENU / SETUP / Cod).

Then enter the SIP URI of the unit to call (see in au-dessus how to deal with special characters and letters), and press .

11.3. Receiving calls

This is very simple, in both cases (direct peer to peer link or SIP server). There is nothing to do…

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, the unit will "follow" the calling unit.

11.4. Links with IP phones

Scoopy+ is compatible with IP phones that use the SIP protocol (many on the market do). The algorithm used in this case is G711, but a few IP phones can also accept G722.

Note that "IP phones" include software SIP phones implemented on computers.

11.5. Additional settings

According to the network, and specially to take into account the jitter the user may modify in order to get the best ratio stable connection/ delay possible on Scoopy+. To access such settings please

Three options are available:

- High : suitable for a good quality and low jitter ; minimum latency but Scoopy+ will be sensitive to potential jitter.
- Medium : intermediate setting, suitable for moderate jitter network ;
- Low : To be selected for highly disturbed network, advise for home ADSL lines. In this case, the transmission is reliable but the latency is higher.

Over a LAN or private network wiht QoS, the High quality setting is recommended as it gives minimum latency.

On the other hand it is highly recommended not to use this setting over the Ethernet[1], as it only supports a low jitter. For example, the user can start with a “medium quality” setting and then decrease to “low quality” if he notices too many audio interferences.

11.6. Notes about the keypad

- Special characters are entered as described in au-dessus.
- To enter lower case characters, continue hitting the key after the number and the upper case letters.
- Once a number or SIP URI has been called, it is easy to recall it without having to type it again: press , then you can scroll through the “history” (last dialled numbers) using the arrows. Press  when the desired number is displayed. This is especially useful for quickly redialling the previous number or URI.

[1] Ceci reste valable même si cette liaison est en mode VPN

12. Maintenance

12.1. Troubleshooting

Power supply failures:

If running on batteries, check that batteries have been inserted properly. Check the “Battery” Green LED indicator on the front panel. The green LED indicates that the battery level is higher than 20%. When the battery LED goes “off”, the remaining autonomy of the Scoopy+ is 15 minutes.

See: Chapiter 3 – Setting up a Scoopy+ - Powering

Note: Replace the old batteries before each new broadcast. Always remove batteries when worn out or when storing the unit for an extended period.

Network Indication:

- Alarm (red)

  “ON” indicates a network problem.
  Check your network.

- Dec (green)

  When "on" indicates that the signal is decoded by Scoopy+.

Unable to establish a connection:

Check the RJ connection between the Scoopy+ and the telephone network. (RJ 11, identified as Tel on the rear panel of Scoopy+ for POTS, and RJ45, identified as ISDN on the rear panel of the Scoopy+ for ISDN)

Connection In ISDN mode

To test your ISDN line, you may connect an ISDN phone or other suitable ISDN verification device into the RJ45 connector instead of the Scoopy+ and call an ISDN number to verify a working ISDN line. Check the ISDN protocol, check the number, and check appropriate setting if going through a PBX.

Connection In POTS mode

To test your POTS line, you may connect a normal phone to the wall connector instead of the Scoopy+ unit and call a normal phone number. Check for proper POTS line settings: Dialing method, «Pulse/Tone", dial tone "Detect/Undetected". Check proper setting if going through a PBX (you may need to dial to get an outside line, Ex 9*).

If the Scoopy+ disconnects while on-line, check for loose connections between the Scoopy+ and the telephone connection. Line noise or interference may be interfering with the modem signals. Retry the connection by dialing the number again.

12.2. Audio section testing

1- Analog section test
a) Connect an audio signal to one of the audio inputs. That signal is available on the headphones (Potentiometer turns that feedback feature off when fully counter-clockwise).

b) Select the menu **< TOOLS >, < MAINTENANCE >, < Test** >: **AD/DA Loop**.

The test is OK if you get the audio signal either on the headphones, or Line out

To end the test go back to the test menu, disable the **AD/DA Loop** by pressing the "none" choice (The star appears on none configuration).

12.3. Network test

These allow checking the network (1) and the remote codec (2)

1- The unit can be configured to loop back to the network the received data.

Select the main menu **< TOOLS >, < MAINTENANCE >, < Test** >: **Loop 2 - Network**.

The loop is enabled as soon as the unit is connected.

2- The unit can be configured to loop back to the network the sent data.

Select the main menu **< TOOLS >, < MAINTENANCE >, < Test** >: **Loop 3 - codec**.

The loop is enabled as soon as the unit is connected.

13. How to open a SCOOPY ready for servicing

Caution: This intervention may cause damages to the unit. It can only be done by authorised person with suitable tools, DES protected.
AAS cannot be responsible for damages caused to person in case of excessive interventions on the unit

Tools required:

Screw Driver (medium-sized)

The Scoopy+ will be now separated into three Sections:
1) The rear metal panel
2) The bonnet
3) The casing lid

Steps :
- Switch to off Scoopy+
- Remove the external mains power cable and batteries
- Undo the two screws at the back.(under the metal rear panel)
- Remove the rear panel
- Undo the four screws under the Scoopy+ (located in wells).
- Remove the bonnet.

Now, you have access to all internal boards.

14. Connectors layout

14.1. POTS Interface

The telephone network connection on Scoopy+ is a RJ 11 connector. (Labeled Analog)

Connector:

Pin	Description
1	-
2	TIP
3	RING
4	-

14.2. ISDN Interface Network

The ISDN connector is a RJ45 – 4 wires into 8 wires

S0/T0 Network :

Point	Description
1	N.C.
2	N.C.
3	TX A, To the network
4	RX A, From the network
5	RX B, From the network
6	TX B, To the network
7	N.C.
8	N.C.

15. Annexes

15.1. ISDN modem information

15.2. ISDN Protocols

ISDN modem supports worldwide ISDN signaling (CCITT I.430, Q.921, Q.931) for voice/audio and data including the following network operator variants :

The ISDN modem supports various international protocols :

- All EuroISDN carriers (Austria, Denmark, Holland, Ireland, Italy, Norway, Portugal, Spain, Switzerland, United Kingdom,...).
- National ISDN-1 and 2 (North America),

15.3. ISDN CLEARING CAUSES

The following table lists the call clearing causes (returned for example in a **CLEARED:** message). Call clearing cause is in hexadecimal. Message meaning is given for an **ETSI ISDN**. Causes with values greater than 80 hex are generated internally.

01 (1)	unallocated (unassigned) number
02 (2)	no route to specified transit network
03 (3)	no route to destination
06 (6)	channel unacceptable
07 (8)	call awarded and being delivered in an established channel

10 (16)	normal call clearing
11 (17)	user busy
12 (18)	no user responding
13 (19)	no answer from user (user alerted)
15 (21)	call rejected
16 (22)	number changed
1A (26)	non-selected user clearing
1B (27)	destination out of order
1C (28)	invalid number format
1D (29)	facility rejected
1E (30)	response to STATUS ENQUIRY
1F (31)	normal, unspecified
22 (34)	no circuit/channel available
26 (38)	network out of order
29 (41)	temporary failure
2A (42)	switching equipment congestion
2B (43)	access information discarded
2C (44)	requested circuit/channel not available
2F (47)	resources unavailable, unspecified
31 (49)	quality of service unavailable
32 (50)	requested facility not subscribed
39 (57)	bearer capability not authorized
3A (58)	bearer capability not presently available
3F (63)	service or option not available, unspecified
41 (65)	bearer capability not implemented
42 (66)	channel type not implemented
45 (69)	requested facility not implemented
46 (70)	only restricted digital information bearer capability is available
4F (79)	service or option not implemented, unspecified

51 (81)	invalid call reference value
52 (82)	identified channel does not exist
53 (83)	a suspended call exists, but this call identity does not
54 (84)	call identity in use
55 (85)	no call suspended
56 (86)	call having the requested call identity has been cleared
58 (88)	incompatible destination
5B (91)	invalid transit network selection
5F (95)	invalid message, unspecified
60 (96)	mandatory information element is missing
61 (97)	message type non-existent or not implemented
62 (98)	message not compatible with call state or message type non-existent or not implemented
63 (99)	information element non-existent or not implemented
64 (100)	invalid information element contents
65 (101)	message not compatible with call state
66 (102)	recovery on timer expiry
6F (111)	protocol error, unspecified
7F (127)	interworking, unspecified
91 (145)	no signaling data link establishment
A2 (162)	no line activation
FF (255)	call clearing, unspecified

15.4. Overview of the SIP protocol

15.4.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

15.4.2. Setting a link with SIP

Let us look at an example (diagram au-dessous): a reporter on the move with a Scoopy+ wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the Scoopy+ is on and connected to the network, it will register itself ❶ to a SIP "registrar". This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar "knows" where the Scoopy+ is, what its IP address is. On the radio house side, a similar process takes place ❷.

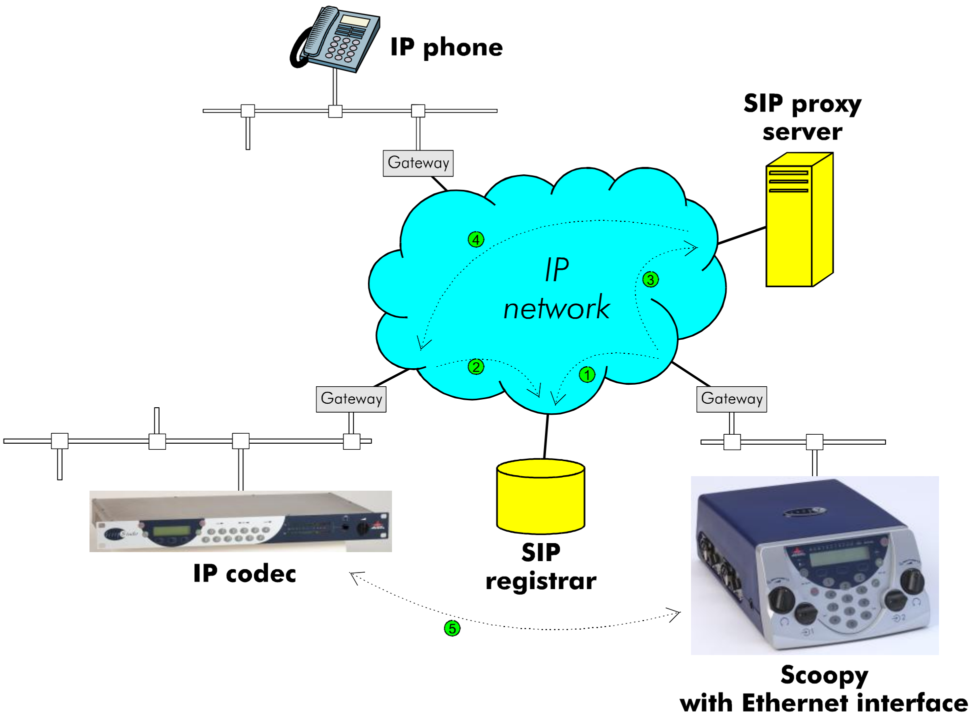
To call the codec of the radio house, the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the Scoopy+ (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

What happens then on the network: the Scoopy+ sends the request ❸ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec "rings"; if it accepts the call, this is notified to the Scoopy+.

At this stage, the proxy(s) provide the Scoopy+ and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ❺ between both units. As a very important feature, the end devices now can exchange data directly; the proxys do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.



15.5. Protocoles de communication supportés dans le mode IP

Le Scoopy+ IP supporte et/ou met en œuvre les protocoles suivants:

- Ethernet (IEEE802.1)
- IP
- DHCP (client)
- HTTP (serveur html intégré)
- Protocoles de transport TCP/IP, UDP/IP
- SIP (Session Initiation Protocol), SDP (Session Description Protocol)
- RTP, RTCP

15.6. Some methods to deal with NAT routers and firewalls

This problem arises when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. One should decide the most suitable solution depending on the specific conditions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges

15.6.1. DMZ

The Scoopy+ can be set in the "DMZ" of the router/firewall. In this way, it is possible to reach directly the Scoopy+ from "outside" the router.

However, such a solution is not recommended for network security reasons, and it should only be used as a temporary test configuration.

15.6.2. Port forwarding

This solution exposes less directly the unit to external attacks. However, it is more complicated, as a number of dedicated ports have to be opened and routed.

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15.6.3. Use of a SIP proxy

A SIP proxy often can deal with the issue. There are various options, e.g.:

- Proxy on the external "public" side: efficient when the router does not block outgoing traffic. Such solution can be implemented e.g. by registering on public SIP servers.
- Proxy inside the LAN area: efficient if the proxy server is allowed to receive and send dedicated traffic. This solution is compatible with restrictive firewalls. Port forwarding has to be set for the proxy to be able to receive calls from outside the LAN.

15.6.4. STUN server

In many cases, a STUN server can be used instead of a dedicated proxy server. STUN is a network protocol allowing a client behind a NAT router (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that might be behind NAT routers.

If a STUN server is available, a Scoopy located behind a NAT router can use the server to complete successfully its call setup with a remote unit outside the NAT router.

15.7. Environnement

Operating temp. Range:	0°C to 45°C (41°F to 113°F)
Humidity:	0 to 90% non -condensing
Storage temp. :	- 20°C to 60°C (-4°F to 140°F)
Dimensions:	(D x W x H) 234 x 155 x 80 mm
Weight:	1.5 kg, with batteries

NOTE